SEISMIC ACQUISITION AND FILTERING

The present invention generally relates to apparatus and
methods for acquiring seismic signal and filtering such
5 data. More particularly, it is directed to a method of
acquiring and processing single sensor recordings,
specifically to derive filters to attenuate the noise in the
thus acquired seismic data.

10   BACKGROUND OF THE INVENTION

Seismic data is collected to analyze the subsurface of the
Earth, and is particularly collected in connection with
hydrocarbon exploration and production activities. Seismic
data for analyzing subsurface structures may be collected on
15 land or over water. In order to obtain seismic data, an
acoustic source is used which typically consists of
explosives or a seismic vibrator on land or an impulse of
compressed air at sea. The seismic signals reflected by the
various geologic layers beneath the surface of the Earth are
20 known as traces and are sensed by a large number, typically
hundreds or thousands, of sensors such as geophones on land
and hydrophones at sea. The reflected signals are recorded
and the results are analyzed to derive an indication of the
geology in the subsurface. Such indications may then be used
25 to assess the likelihood and location of potential
hydrocarbon deposits.

Seismic surveys are generally conducted using one or more
receiver lines having a plurality of receiver station
30 locations spaced evenly along their lengths. In a two
dimensional (2D) survey, a single receiver line is used and
the acoustic source is typically positioned at various

points in-line with the receiver line. In a three
dimensional survey, a plurality of parallel receiver lines
are typically used and the acoustic source is generally
positioned at various points offset from the receiver lines.

5   While a 2D seismic survey can only create a cross-sectional
representation of the subsurface, a 3D seismic survey can be
used to develop a three dimensional representation of the
subsurface.

10   Seismic data are subject to a wide variety of noise related
problems that can and do limit its usefulness. Broadly
speaking, noise found in seismic traces is either incoherent
or coherent. Incoherent ambient noise, or uncorrelated
"white" noise, is ubiquitous and is generally greatly

15   attenuated through the simple expedient of stacking,
although extremely large individual data values ("spikes")
and "bad" traces often need special attention. Coherent, or
correlated, noise on the other hand cannot usually be so
readily eliminated. Some common examples of coherent noise

20   include multiple reflections, ground roll, air waves, guided
waves, sideswipe, cable noise and 60 hertz power line noise.
Among the many known approaches to attenuating noise, there
are space-time based or transform based methods. Space-time
based methods operate on the time series of the acquired

25   data. Transform based methods operate on data transformed
from the space-time domain into another domain using a
suitable transformation. The most popular of the 2-D
transform methods is the 2-D Fourier transform (or "f-k"
transform). Seismic data containing noise are transformed to

30   the alternative domain where noise events are more compactly
represented. If the noise energy can be located and isolated
in the transform domain, it is removed from the transformed

data by filtering or muting, i.e. by setting the values in
the region corresponding to the noise energy equal to zero
or some other minimal value. Finally, the transformed data,
without the noise energy, are then inverse transformed to
5   return them to the time and offset (i.e., untransformed or
"x-t") domain.

In conventional seismic data acquisition systems data are
inherently filtered through use of "hard-wired"
10   (electrically connected) groups of sensors. A group or
receiver array delivers a single output trace (the
normalized sum or arithmetic average of the output of all
individual sensors of the group) at the particular receiver
station location about which the sensors are placed. The
15   single trace is the normalized sum or arithmetic average of
the output of all individual sensors making up the group.
Without further processing, such a two-dimensional group has
a spectral response that can be approximated by a frequency-
independent 2D sinc function in the wavenumber or kx-ky
20   domain.

More recently, however, seismic surveys have been performed
using single or point receiver arrays. Such surveys offer
the potential of recording the output of individual sensors
25   or receivers and the inherent filtering effect of the hard-
wired group can be replaced by filters that are better
adapted to the nature of seismic noise and preserve more of
the seismic reflection signals.

30   It is therefore an object of the present invention to
provide methods for processing seismic data, particularly
methods for designing and applying filters for such data.

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SUMMARY OF THE INVENTION

The invention includes a method of determing a digital
filter for seismic signals comprising the steps of: defining
contraints representing a filter for preserving signals
representing reflection and/or refractions from sub-surface
structure and suppressing noise signals in seismic signals;
and using an iterative process with each iteration
comprising: transforming a filter obtained from a previous
iteration into a transform domain; applying in said
transform domain first constraints; inverse transforming the
filter with the applied constraints into a sample domain;
and applying in said sample domain second constraints to
obtain an iterated filter.

The term sample domain and transform domain are arbitrary in
the sense that both domain are representations of the seimic
signals using a different set of coordinates. The transform
describes the change from one domain to the other and the
inverse transform describes the reverse direction.
Preferably the transform is between the wavenumber or
frequency-wavenumber domain and and the spatial or temporal-
spatial domain. This transform can be performed by thewell-
known Fourier transform.

In a preferred embodiment the filter is constrained to a
predefined tolerance in one domain and to a predefined
response outside a finite region in the other. The
predefined response is in most case very small or zero to
suppress the signal outside the finite region.

The method is preferable applied using a non-rectangular
transform which can be described as staggered or hexagonal,
even though the hexagonal case is only a special case of the
general staggered transform used.

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Application of this staggered transform yields superior
results when applied to data acquired on a staggered grid.
It is therefore another aspect of the ibnvention to use
groups of receivers or single sensor seismic receivers

10   distributed to obtain seismic measurements on a staggered or
hexagonal grid. A staggered grid can be characterised by
having non-diagonal elements in the matrix that translates
the grid points onto themselves. In a more practical
definition a staggered grid can be seen as an array of

15   points in which every second rows of of points are shifted
with respect to the remaining rows or as two overlapping and
shifted rectangular grids.

The filter of the present inventions are preferably is a

20   zero-phase finite impulse response (FIR) filter. They have
at least at least two dimensions, but are preferably full
three-dimensional (3D) filters.

These and other aspects of the invention will be apparent

25   from the following detailed description of non-limitative
examples and drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIGs. 1A-D show a desired ideal wavenumber (kx-ky) response

30   of a 2D x-y filter in three different cross-sections;

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FIG. 2 illustrates steps to design a filter in accordance with an example of the invention;

5   Fig. 3 shows the wavenumber (kx-ky) response of a 2D x-y filter designed in accordance to the method of FIG. 2 using a rectangular discrete fast Fourier transform;

Fig. 4 shows the wavenumber (kx-ky) response of a 2D x-y filter designed in accordance to the method of FIG. 2 using
10   a hexagonal discrete fast Fourier transform; and

FIGs. 5A,B compare single receiver data sets obtained by processing in accordance with an example of the present invention (FIG. 5B) with a data set obtained with
15   conventional hard-wired groups and conventional noise filtering (FIG. 5A) after differential moveout (DMO) stacking.

DETAILED DESCRIPTION

20   A filter in accordance with the present invention is designed to remove at least part of the noise from the data acquired through a seismic survey. The known noise characteristics and seismic signal spectrum depend on the various parameter such as source and receiver locations, the
25   acoustic properties of near surface layers and many other parameters within or outside the control of an operator. Any such pre-established knowledge of the noise characteristics can be used to establish a set of filter parameters that when combined provide a description of the filter suitable
30   to be used as an input to a data processing machine or computer.

The example of FIG. 1 shows the ideal wavenumber response of
a 2-D filter. Two schematic perpendicular cross-sections are
shown in FIG. 1A illustrating parameter for a functional
description of the design in the wavenumber or (f - kx - ky)
5   domain. FIG. 1B shows the f - kx cross-section for ky = 0,
FIG. 1C shows the middle panel shows the f - ky cross-
section for kx = 0, and FIG. 1D shows the kx - ky cross-
section at f = 10 Hz.

10   In the 2-D survey of this example, most of the reflection
energy is expected to be incident in, or near the in-line
plane, whereas most of the energy incident in the cross-line
direction should be scattered ground-roll and other types of
noise. The passband of the initial 2-D x - y filter is
15   therefore designed with elliptical contours, with greater
attenuation in the cross-line direction. The larger passband
in the in-line direction is designed to preserve signal
components with maximum wavenumber and frequency content.

20   The filter of FIG. 1 is characterized by three frequency
bands separated by the frequencies fbreak1 and fbreak2.
Within each frequency band there is defined a pass zone (the
central zone), a transition zone and an outer reject or stop
zone. The boundaries of the zone are marked by piecewise
25   linear function defined through the slowness parameters s1
to s6.

Further parameters characterizing the filter of the example
is the aspect ratio of the elliptical pass zone and the
30   desired attenuation in the stop band and the width of the
transition zone.

Other parameter define the dimensions (length, width) of the
filter or the filter support for a finite impulse response
filter (FIR) as used in this example.

5   Filters can be characterized by parameters others than those
described above. But independent of the specific form of
representation, it is always possible to define a
representation of a desired filter in machine readable form.

10 Each of the parameters of such an ideal filter can be
regarded as constraints in the time domain and in the
frequency domain and the problem of designing a real filter
is an approximation optimized with those constraints.

15 It is known that if the constraints define convex sets in
the set of square summable sequences, then applying
alternating orthogonal projections onto these would converge
to the optimal solution. This is called the method of
Alternating Projections onto Convex Sets (APOCS). The APOCS
20 method has been used in various technical fields outside
seismic processing. For details of the APOCS method
reference is made to Çetin et al. (IEEE Signal Procesing
Magazine, vol. 14 (1997), pp.60-64 or Proc. Int. Symp.
Circuits and Systems, 1987, pp. 1044-1047) who adapted the
25 APOCS method to the design of zero-phase 2-D finite impulse
response (FIR) filters.

In the APOCS method the filter design problem is formulated
30 to alternately satisfy the frequency domain constraints on
the magnitude response bounds and sample (space and/or time)
domain constraints on the impulse response support. The

algorithm is iterative and each iteration requires two 2-D
FFT computations. Çetin *et al.* (see above) have shown that if
convergence is achieved, this approach results in filters
with approximately equiripple response.

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It can be proven that the APOCS method is globally
convergent under certain conditions. If the constraints in
the sample and spectral domains define convex sets in the
set of square summable sequences, then the imposition of the

10   constraints in the sample and spectral domains are the
orthogonal projections onto these sets, and if the sets
intersect, the iterates converge to a member in the
intersection set. Furthermore, if there is just one sequence
satisfying both conditions, then this sequence is the

15   equiripple solution of the filter design problem.
Consequently, the iterations converge to the equiripple
filter.

Generalizing to the design of a 3-D filter, the filter is to

20   be zero-phase, and its spectral response in the f-kx-ky
domain, $H(f,k_x,k_y)$ to be within given tolerance ranges:

$$[1] \quad \begin{aligned} H_{id}\left(f,k_x,k_y\right) - E_d\left(f,k_x,k_y\right) &\le H_{id}\left(f,k_x,k_y\right) \\ &\le H_{id}\left(f,k_x,k_y\right) + E_d\left(f,k_x,k_y\right), \end{aligned}$$

25   where $H_{id}(f,k_x,k_y)$ is the ideal filter response, and
$E_d(f,k_x,k_y)$ is the tolerance, or the desired maximum ripple
level, which may take different values in different
passbands and stopbands. The ideal response is defined as

30    $$[2] \quad H_{id}\left(f,k_x,k_y\right) = \begin{cases} 1, & \text{if } \left(f,k_x,k_y\right) \in F_P, \\ 0, & \text{if } \left(f,k_x,k_y\right) \in F_S, \end{cases}$$

and the tolerances as

$$[3] \qquad E_d\left(f, k_x, k_y\right) = \begin{cases} \delta_P, & \text{if } \left(f, k_x, k_y\right) \in F_P, \\ \delta_S, & \text{if } \left(f, k_x, k_y\right) \in F_S, \end{cases}$$

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where F_p and F_s are the pass and stop bands, and δ_P and δ_S are the corresponding tolerances, respectively.

In the t-x-y domain, the filter has a finite-extent support,
10   I, which is a symmetric region around the origin in order to have a zero-phase response. The time-space domain constraint requires that the filter coefficients are zero outside the region I.

15   The iterative method is initialized with an arbitrary finite-extent, real 3-D sequence that is symmetric around the origin in order to have a zero-phase response. The time/space domain constraint requires that the filter coefficients must be equal to zero outside the region I. As
20   an initial estimate, the inverse Fourier Transform (F^{-1}) of the ideal frequency response is used:

$$[4] \qquad h_{id}(t, x, y) = F^{-1}\{H_{id}(f, k_x, k_y)\}$$

25   with

$$[5] \qquad h_0\left(t, x, y\right) = \begin{cases} h_{id}\left(t, x, y\right), & \text{if } \left(t, x, y\right) \in I, \\ 0, & \text{otherwise.} \end{cases}$$

FIG. 2 illustrated the iterative process beginning with the
30   selection of an initial time domain response h_{id}.

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At each iteration of the algorithm, the time/space and
frequency/wavenumber domain constraints are successively
imposed onto the current iterate. The n-th iteration

5   consists of the following steps:

- Compute the 3D Fourier Transform of the n-th iterate
$h_n(t,x,y)$ by a 3D FFT algorithm to obtain $H_n(f,kx,ky)$.

- Impose the frequency/wavenumber domain constraints as:

$$G_n\left(f,k_x,k_y\right) =$$

$$
\begin{cases}
H_{id}\left(f,k_x,k_y\right) + E_d\left(f,k_x,k_y\right) \\
\quad \text{if } H_n\left(f,k_x,k_y\right) > H_{id}\left(f,k_x,k_y\right) + E_d\left(f,k_x,k_y\right) \\
\\
H_{id}\left(f,k_x,k_y\right) - E_d\left(f,k_x,k_y\right) \\
\quad \text{if } H_n\left(f,k_x,k_y\right) < H_{id}\left(f,k_x,k_y\right) + E_d\left(f,k_x,k_y\right) \\
\\
\qquad H_n\left(f,k_x,k_y\right) \text{ otherwise.}
\end{cases}
$$

10   [6]

- Compute the 3-D inverse Fourier Transform of $G_n(f,kx,ky)$
to obtain $g_n(t,x,y)$.

- equal $g_n(t,x,y)$ to zero outside the region I to obtain
$h_{n+1}(t,x,y)$ as

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[7] $$h_{n+1}\left(t,x,y\right) = \begin{cases} g_n\left(t,x,y\right), & \text{if } \left(t,x,y\right) \in I, \\ 0, & \text{otherwise.} \end{cases}$$

- If the mean-squared error between the iterates $h_n(t,x,y)$
and $h_{n+1}(t,x,y)$ is less than a predefined threshold, then
exit.

20

These steps are performed using a standard numerical programme such as Matlab(TM) and the routines embedded therein.

5   The method outlined above can be directly applied to seismic data acquired on a rectangular grid by using a 2D rectangular fast Fourier transformation (FFT) that samples the data at rectangularly distributed points.

10   However, the advent of single sensor recording makes it more feasible to acquire seismic data on a staggered layout of receivers. It is known that circularly bandlimited signals require 13 percent less samples to achieve the same spectral resolution when sampled on a staggered grid than being

15   sampled on a rectangular grid. Since seismic signals and noise can be assumed to be circularly bandlimited in the wavenumber space, hexagonal sampling is more efficient than rectangular sampling potentially reducing the number of receivers in a survey by 13 percent.

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It is therefore advantageous to adapt the present method to accommodate surveys performed using a hexagonal distribution of receivers or groups of receivers.

25   It is feasible to apply the above method to data acquired on a staggered or hexagonal grid by using the following steps:
    -spatial oversampling:
    -performance of a rectangular FFT;
    - imposing spectral constraints in the wavenumber domain;
30   - using a rectangular inverse Fourier transform to revert to the time domain; and

- imposing zeros on points that do not correspond to
receiver locations.

FIG. 3 shows the resulting response of a 2D filter with dB

5   contour levels for a hexagonal receiver grid using a
rectangular FFT as outlined in the previous paragraph. As in
FIG. 1, the top panel shows the f - kx cross-section for ky
= 0, the middle panel shows the f - ky cross-section for kx
= 0, and the bottom panel shows the kx - ky cross-section at

10   f = 10 Hz. Comparison with FIG. 1 shows that signals from
outside the desired region (sidelobes) leak into the
filtered data.

This is a consequence of a mismatch between the true

15   receiver locations which for this example were placed on a
hexagonal grid and the rectangular DFT used for the
iterative process. The method introduces holes into the
sample domain, which is no longer convex. Convergence is
then no longer guaranteed and the above filter design

20   method is not expected to give optimum results.

It was found that the performance of the filter design
method can be significantly improved by using a Fourier
transform with hexagonal sampling in the space (x-y) domain

25   when the data is acquired on a hexagonal grid.

A discrete Fourier transform (DFT) that relates a hexagonal
sampled signal to hexagonal sample of its Fourier transform
is known as such and is given for example by

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[8]

$$X(k_1, k_2) = \sum_{n1=0}^{3N-1} \sum_{n2=0}^{N-1} x(n_1, n_2) \exp\left[-j\frac{2\pi}{3N}\{(2n_1 - n_2)k_1 + 3n_2k_2\}\right]$$

5   For this and further details of the hexagonal DFT reference
can be made for example to R. M. Mersereau and D. E Dudgeon
in "Multidimensional Digital Signal Processing", Prentice-
Hall 1984, 97-100.

10   FIG. 4 shows the response resulting from the use of a
hexagonal DTF of the 2D filter with dB contour levels. As in
FIG. 1, the top panel shows the f - kx cross-section for ky
= 0, the middle panel shows the f - ky cross-section for kx
= 0, and the bottom panel shows the kx - ky cross-section at
15   f = 10 Hz. Comparison with FIG. 1 shows a very good match
between the desired and the resulting responses, and the
sidelobe levels of the stop band are less than 35 dB
compared to the higher sidelobes apparent in FIG. 3.

20   It was found that the filter design results in improved data
processing, particularly when used on seismic data acquired
through single sensor surveys. To illustrate the improvement
the same data set was acquired using (a) conventional groups
of receivers, (b) single sensors and (c) a very low density
25   of single sensors. After filtering the data was further
processed using a stacking procedure with differential move-
out (DMO) correction.

FIG. 5A shows the DMO stack obtained using conventional
30   filtering methods on signals from analogue groups of 72
receivers per 30 metres, while FIG. 5B depicts the DMO stack

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obtained after processing by an adaptive pre-filter and an
2D filter designed through the iterative process described
above (FIG. 2), using 36 receivers per 30 metres. Compared
to the analogue result, FIG. 5B demonstrates considerable
5   increase in S/N, especially for higher CMP's.